Atty Docket No. 6655P016

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE
BEFORE THE BOARD OF PATENT APPEALS AND INTERFERENCES

In re Application of:

Khosrow Lashkari, et al

Application No.: 10/023,826

Filed: December 19, 2001

For: JOINT OPTIMIZATION OF SPEECH EXCITATION AND FILTER PARAMETERS

Examiner: David D. Knepper

Art Group: 2654

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APPEAL BRIEF UNDER 37 C.F.R. § 41.37(a)

This is an appeal to the Board of Patent Appeals and Interferences from the decision of the Examiner of Group 2654, dated January 27, 2006, which finally rejected claims 1-25in the above-identified application. This Appeal Brief is hereby submitted pursuant to 37 C.F.R. § 41.37(a).

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I. REAL PARTY IN INTEREST

The real party in interest is the assignee of the full interest in the invention as claimed, NTT Docomo, Inc., a Japanese Corporation having a principle address at Sanno Park Tower, 11-1, Nagata-cho, 2-chome, Chiyoda-ku, Tokyo, Japan 100-6150.

II. RELATED APPEALS AND INTERFERENCES

To the best of Appellant's knowledge, there are no appeals or interferences related to the present appeal that will directly affect, be directly affected by, or have a bearing on the Board's decision in the instant appeal.

III. STATUS OF THE CLAIMS

Claims 1-25 are pending in the application and were finally rejected in an Office Action mailed January 27, 2006. Claims 1-25 are the subject of this appeal. A copy of Claims 1-25 as they stand on appeal are set forth in Appendix A.

IV. STATUS OF AMENDMENTS

No amendments have been submitted subsequent to the Final Office Action mailed March 13, 2006.

V. SUMMARY OF CLAIMED SUBJECT MATTER

Appellant's invention as claimed in claims 1-25 is directed to methods and a system digitally encoding speech. The system may be an analysis-by-synthesis (AbS) system, which is also referred to as a source-filter model. Source-filter models are designed to mathematically model human speech production. An efficient optimization algorithm is used for multipulse speech coding. The efficient optimization algorithm performs calculations using contributions of the non-zero pulses of the excitation function and not the zeros of the excitation function. The efficient optimization algorithm is used to minimize the synthesis error between the original speech signal and the synthesized speech signal.

Independent claim 1 claims a method of digitally encoding speech, including: generating an excitation function using an excitation module, where the excitation function comprises a number of non-zero pulses within an analysis frame separated by spaces therebetween (e.g., Figs. 1, 2A, 2B, and 3; pp. 5-7 and 9-11), generating synthesized speech using a synthesis filter from the number of non-zero pulses within the analysis frame without contribution from the spaces therebetween (e.g., Figs. 1, 2A, 2B, and 3; pp. 9-13), performing synthesis filter optimization, including selecting one of a plurality of excitation functions and selecting roots of the synthesis polynomial for one excitation function that

minimizes a synthesis error produced by the synthesis filter (e.g., Figs. 1, 2A, 2B, and 3; pp. 9, and 11-15).

Independent claim 16 claims a method of digitally encoding speech, including: producing a series of pulses within an analysis frame, adjacent pulses defining a space therebetween (e.g., Figs. 1, 2A, 2B, and 3; pp. 5-13), and generating a synthesis polynomial, where generating the synthesis polynomial comprises calculating a contribution of the pulses and not calculating a contribution of only the space (e.g., Figs. 1, 2A, 2B, and 3; pp. 9-13), and including selecting one of a plurality of excitation functions and selecting roots of the synthesis polynomial for the one excitation function that minimizes a synthesis error produced by the synthesis filter (e.g., Figs. 1, 2A, 2B, and 3; pp. 9, and 11-15).

Independent claim 19 claims a speech synthesis system, including an excitation module responsive to an original speech and generating an excitation function using an excitation module, where the excitation function comprises a series of pulses within an analysis frame (e.g., Figs. 1, 2A, 2B, and 3; pp. 5-7 and 9-11), and a synthesis filter responsive to the excitation function and the original speech and generating a synthesized speech using a synthesis filter (e.g., Figs. 1, 2A, 2B, and 3; pp. 9-13); wherein the synthesis filter computes a convolution of an impulse response and the excitation function, and the convolution computation comprises calculating samples of speech having only the pulses within the analysis frame (e.g., Figs. 1, 2A, 2B, and 3; pp. 5-7 and 9-11); including selecting one of a plurality of excitation functions and selecting roots of the synthesis polynomial for the one excitation function that minimizes a synthesis error produced by the synthesis filter (e.g., Figs. 1, 2A, 2B, and 3; pp. 9, and 11-15).

VI. GROUNDS OF REJECTIONS TO BE REVIEWED ON APPEAL

A. Whether claims 1-25 comply with the written description requirement under 35 U.S.C. §112, first paragraph.

B. Whether claims 1-25 are indefinite for failing to particularly point out and distinctly claim the subject matter which applicant regards as the invention under 35 U.S.C. §112, second paragraph.

C. Whether claims 1, 3-7 and 12 are patentable over U.S. Patent No. 5,664,055 of Kroon et al. ("Kroon") in view of U.S. Patent No. 6,449,590 of Gao et al. ("Gao").

D. Whether claim 2 is patentable over Kroon in view of Gao.

E. Whether claim 8 is patentable over Kroon in view of Gao.

F. Whether claim 9-11 and 13-15 are patentable over Kroon in view of Gao as applied to claims 1-8, 12, 16, 17, and 19 in further view of "A New Algorithm for Parameter Re-Optimization in Multi-Pulse Excitation LP Synthesizer" by Chen ("Chen").

G. Whether claim 16 is patentable over Kroon in view of Gao.

H. Whether claim 17 is patentable over Kroon in view of Gao.

I. Whether claim 18 is patentable over Kroon in view of Gao as applied to claims 1-8, 12, 16, 17, and 19 in further view of Chen.

I. Whether claim 19 is patentable over Kroon in view of Gao.

K. Whether claims 20-25 are patentable over Kroon in view of Gao as applied to claims 1-8, 12, 16, 17, and 19 in further view of Chen.

VII. ARGUMENT

The claims do not stand or fall together.

A. Claims 1-25 comply with the written description requirement.

Initially, the Office action rejected claims 2 and 15 under 35 U.S.C. 5 112, first paragraph, as failing to comply with the written description requirement. In response to the Appellants arguments of the Response to Office action, mailed October 19, 2005, the Office action rejected claims 1-25 under 35 U.S.C. 5 112, first paragraph, as failing to comply with the written description requirement. See Office action, mailed January 27, 2006, page 3. Specifically, the Office action states that joint optimization is not clearly taught by the Appellant. Id. Appellant respectfully disagrees. Joint optimization is taught. Essentially, excitation function and roots of the synthesis polynomial for the excitation function are selected to reduce the synthesis error produced by the synthesis filter. As described, the selection of these two items is performed to obtain the optimum combination of excitation function and roots of the synthesis polynomial. The manner in which the excitation function and roots of the synthesis function are selected is clearly described in the specification. For example, see page 9, line 13 to page 17, line 24.

For example, Figure 2B is a flow chart of an alternative speech synthesis system using joint optimization of the model parameters and the excitation signal. The description corresponding to Figure 2B discloses the polynomial coefficients are used to find the *optimum excitation function* u(n) from a codebook, and that the *polynomial coefficients are optimized.* See page 16, lines 14-30. To make optimization of the coefficients easier, the

polynomial coefficients are first converted to the roots of the polynomial A(z) and a gradient search algorithm is used to optimize the roots. See id. Accordingly, optimization of both the model parameters and the excitation signal are described in the specification.

The Office action also contends that specification is not directed to joint optimization, stating that the improvement lies only in using the optimization algorithm to reduce the computational load required to compute the synthesized speech s(n) by taking into account the sparse nature of the excitation pulses. See Office action, mailed January 27, 2006, citing the specification at page 17, lines 16-24. Although the Office action correctly cites the specification that the improved optimization algorithm reduces the computational load, the Office action fails to recognize that this particular improvement is described in the context of only one of the optimizations described in the specification. As described above, the specification clearly discloses optimizing both excitation function and the roots of the synthesis polynomial for the excitation function to reduce the synthesis error produced by the synthesis filter.

In addition, the Office action contends that the Appellant "may be trying to claim parallel processing whereon only series processing steps are disclosed (i.e. – figure 3)." See Office action, mailed January 27, 2006. Appellant respectfully submits that regardless of whether the processing is done in series or in parallel, the contention of the Office action is misplaced because the Office action is interpreting joint optimization to equate to simultaneous optimization, or that the optimization for both the model parameters and excitation function are performed in parallel. Accordingly, Appellant respectfully submits that whether simultaneous or parallel optimization is disclosed or not, the specification clearly discloses joint optimization, and thus, complies with the written description.

For the forgoing reasons, the Appellant respectfully submits that joint optimization is clearly taught by the Appellant to comply with the written description requirement. Accordingly the Appellant respectfully submits that the rejections to claims 1-25 under 35 U.S.C. § 112, first paragraph, be withdrawn.

As for the initial rejections of the original claims, claims 2 and 15, under 35 U.S.C. § 112, first paragraph, the Appellants respectfully submit that the examiner has the initial burden, after a thorough reading and evaluation of the content of the application, of presenting evidence or reasons why a person skilled in the art would not recognize that the written description of the invention provides support for the claims, and that there is a strong presumption that an adequate written description of the claimed invention is present in the specification as filed. See Manual of Patent Examining Procedure ("M.P.E.P.") §2163(II)(A), citing *Wertheim*, 541 F.2d at 262, 191 USPQ at 96 (which indicates that the rejection of an original claim for lack of written description should be rare. The Examiner in the Office action, mailed July 28, 2005, contended that the claims contain subject matter which was not described in the specification in such a way as to reasonably convey to one skilled in the relevant are that the inventor(s), at the time the application was filed, had possession of the claimed invention only because the specification on page 13 indicates that a standard root finding algorithm is utilized, indicating that the subject matter of claim 2 and 15 is from another and is not the applicant's invention. See Office action, mailed July 28, 2005, page 3. As described in the response to the Office action, mailed July 28, 2006, the Appellants agree with the Examiner that the standard root finding algorithm at page 13 is not novel, however, claims 2 and 15 set forth an iterative root optimization algorithm, and not a standard root finding algorithm. For example, the root optimization algorithm may include a

standard root finding algorithm to find the roots of the synthesis filter polynomial A(z). See operation 56 of Figure 3, and page 17, lines 10-12. Next, the roots of the synthesis polynomial *are optimized* with an iterative gradient search algorithm using formulas (27), (25), (24) and (23). See operation 58 of Figure 3, and pages 17, lines 12-14. In other words, finding the roots of the polynomial using the standard root finding algorithm may be performed as a separate operation before optimizing the roots of a synthesis filter polynomial, which uses an iterative root optimization algorithm, not the standard root finding algorithm. Accordingly, the specification clearly describes "optimizing roots of the synthesis filter polynomial using an iterative root algorithm", regardless of whether or not it uses a standard root finding algorithm to find the roots before the roots are optimized.

Furthermore, even if the iterative root optimization algorithm includes common iterative solutions (e.g., common mathematical solutions) for root finding, the Examiner has the initial burden of presenting evidence or reasons why a person skilled in the art would <u>not</u> recognize that the written description of the invention provides support for the claims. The Office action has merely indicated that since the iterative root optimization algorithm is iterative, it is admitted prior art since iterative solutions are the most common mathematical solutions for root finding problems. This assertion by the Office action actually strengthens that fact that a person skilled in the art would recognize that the written description of the invention provides support for the claims.

For the forgoing reasons, the Appellant respectfully submits that optimizing roots of a synthesis filter polynomial using an iterative root optimization algorithm is clearly taught by the Appellant to comply with the written description requirement. Accordingly the Appellant

respectfully submits that the rejections to claims 1-25 under 35 U.S.C. § 112, first paragraph, be withdrawn.

B. Claims 1-25 particularly point and distinctly claim the subject matter which Appellant regards as the invention.

The Office action has rejected claims 1-25 under 35 U.S.C. § 112, second paragraph. Appellant amended the claims to overcome the rejection. In particular, the term "computing" was changed to "generating," and the operations of "selecting one of a plurality of excitation functions and selecting roots of the synthesis polynomial for one excitation function that minimizes a synthesis error produced by the synthesis filter" were amended to clarify that they are part of the limitation of "performing synthesis filter optimization."

The Office action contends that claim language of "selecting one of a plurality of excitation functions and selecting roots" implies a look up table, and that by adding this language to the "computing" step of the method is contradictory over the limitation that "computing" is performed in response to only said number of non-zero pulses, which indicate nothing expect for the non-zero pulses can be used. This contention, however, is moot in view of the amended claims presented in the Appellant's response, mailed March 13, 2006, which amended claim 1 to clarify preexisting claim limitations, as described above.

Furthermore, the amendments made to claim 1 by the Appellant in the Appellant's response, mailed March 13, 2006, further clarify the separate operations of generating an excitation function, generating synthesized speech, and performing synthesis filter optimizations.

For the foregoing reasons, Appellant respectfully submits that the claims 1-25 particularly point out and distinctly claim the subject matter which the Appellant regards as the invention. Accordingly, the Appellant respectfully submits that the rejections of claims 1-25 under 35 U.S.C. § 112, second paragraph, be withdrawn.

C. Claims 1, 3-7 and 12 are patentable over Kroon in view of Gao.

As described above, independent claim 1 includes limitations of generating an excitation function using an excitation module, where the excitation function comprises a number of non-zero pulses within an analysis frame separated by spaces therebetween; generating synthesized speech using a synthesis filter from the number of non-zero pulses within the analysis frame without contribution from the spaces therebetween; and performing synthesis filter optimization, including selecting one of a plurality of excitation functions and selecting roots of the synthesis polynomial for one excitation function that minimizes a synthesis error produced by the synthesis filter.

To establish *prima facie* obviousness of a claimed invention, all the claim limitations must be taught or suggested by the prior art. See M.P.E.P. §2143.03, citing *In re Royka*, 490 F.2d 981, 180, USPQ 580 (CCPA 1974). Appellants respectfully submit that neither Kroon nor Gao, alone or in combination, disclose all the limitations of claim 1.

First, Appellant respectfully submits that claim 1 requires the operation of generating an excitation function using an excitation module, where the excitation function comprises a number of non-zero pulses within an analysis frame separated by spaces therebetween. Neither Kroon nor Gao, alone or in combination, disclose this limitation of the claim.

The Office action contends that Kroon discloses this limitation in Figure 1 of Kroon, stating that "Figure 1 shows pulses upon which computations must be made in combination with spaces which do not require any computations and is used by Kroon as background showing how earlier systems (fig. 2) used a simpler pattern of excitation pulses which merely repeated. See Office action, mailed January 27, 2006, pages 5-6. However, Appellant respectfully submits that nothing in Kroon discloses generating an excitation function using an excitation module, where the excitation function comprises a number of non-zero pulses within an analysis frame separated by spaces therebetween."

Kroon is directed to a decoder for voiced/unvoiced classification of speech for excitation codebook selection in CELP speech decoding during frame erasures. See Kroon, Title and Abstract. Kroon discloses that an excitation signal is generated, however, this excitation signal does not include a number of non-zero pulses separated by spaces therebetween in an analysis frame. Kroon discloses that decoder generates an excitation signal based on a classification whether a speech signal is periodic (i.e., voiced) or non-periodic (i.e., unvoiced). See col. 2, lines 54-60. More specifically, when the speech signal is classified as periodic, the excitation signal is generated based on the output of a first portion (e.g., active codebook (ACB)) of the decoder, and not on the output signal of a second portion (e.g., active codebook (ACB)) of the decoder. Conversely, when the speech signal is classified as non-periodic, the excitation signal is generated based on the output of the second portion, and not on the output signal of the first portion. See col. 2, lines 61-67. Consequently, Kroon does not disclose that the excitation signal includes a number of non-zero pulses separated by spaces therebetween, but only that the excitation signal is either based on the output signal of either the first or second portions of the decoder.

Gao fails to cure the deficiency described above with respect to Kroon. Gao is directed to a speech encoder using warping in long term preprocessing. See Gao, Abstract. The encoder of Gao, to support lower bit rate encoding modes, may classify the input signal as either voiced or unvoiced speech, similarly to the decoder described in Kroon. See Figure 2 and col. 6, lines 5-39. In particular, the speech encoding circuitry simultaneously uses both the adaptive and fixed codebook (ACB and FCB) excitation vectors (i.e., signals) to minimize a third error signal. See col. 6, lines 40-45. Gao also discloses that the excitation signal for an LPC synthesis filter is built from the two traditional components: 1) the pitch contribution; and 2) the innovation contribution. The pitch contribution is provided through use of an adaptive codebook. An innovation or fixed codebook has several subcodebooks in order to provide robustness against a wide range of input signals. To each of the two contributions a gain is applied which, multiplied with their respective codebook vectors and summed, to provide the excitation signal. See col. 40, lines 52-60. Consequently, Gao does not disclose that the excitation signal includes a number of non-zero pulses separated by spaces therebetween, but only that the excitation signal is the summation of the excitation vectors from the ACB and the FCB. Accordingly, Gao fails to cure the deficiency noted above with respect to Kroon.

Therefore, neither Kroon nor Gao, alone or in combination, disclose this limitation of the claim.

Second, Appellant respectfully submits that claim 1 requires the operation of "performing synthesis filter optimization, including selecting one of a plurality of excitation functions and selecting roots of the synthesis polynomial for one excitation function that

minimizes a synthesis error produced by the synthesis filter." Neither Kroon nor Gao, alone or in combination, disclose this limitation of the claim.

The Office action asserted that it was difficult to tell whether all claim elements merit consideration because of the confusion noted above with respect to the rejections under 35 U.S.C. §112, second paragraph. Consequently, the Office action asserted Gao to meet this limitation. Appellant respectfully submits, and the Office action concedes that Kroon fails to disclose this limitation of the claim. Gao, however, fails to cure the deficiency of Kroon.

Gao does disclose a joint optimization; however, the joint optimization of Gao, as indicated by blocks 307 and 309 of Figure 3, identifies the optimum gain for the excitation signals output from the adaptive and fixed codebooks (ACB and FCB). This is done by generating a synthesized and weighted signal via block 301 and 303, that best matches the first target signal 229, which minimizes the third error signal. In other words, Gao only discloses joint optimization of the gains for both the adaptive and fixed codebooks vector selections to minimize the error signal.

The Office action also contends that this limitation is met because Gao discloses that the use of Line Spectral Pair (LSP) inherently performs transformation of LPC parameters which are roots of these polynomials on the z-unit circle. See Office action, mailed January 27, page 6. Gao does in fact discloses that the LSP (which are also called Line Spectral Frequencies) is a transformation of LPC parameters, and that the LSPs are obtained by decomposing the inverse filter transfer function A(z) to a set of two transfer functions, one having even symmetry and the other having odd symmetry. The resulting LSPs are the roots of the two transfer functions. See col. 47, lines 57-64. However, Gao also discloses that the encoder analyzes each speech signal to extract the parameters of the CELP model, i.e., the

LP filter coefficients, (LPC parameters) adaptive and fixed codebook indices and gains. These parameters are encoded and transmitted every subframe of the speech frame. In addition, LP analysis at the block 239 is performed twice per frame *but only a single set of LP parameters is converted to line spectrum frequencies (LSF) and vector quantized* using predictive multi-stage quantization (PMVQ). See col. 8, line 53 to col. 9, line 6. Gao, however, does not disclose that the roots or LSFs are used to perform synthesis filter optimization. The optimization of the gains of the ACB and the FCB specifically refer to using the LPC coefficients. See col. 6, lines 5-39, specifically lines 14-17. Accordingly, Gao only discloses that the synthesis filter optimization is performed using LPC coefficients, and not by selecting roots of the synthesis polynomial for one excitation function that minimizes a synthesis error produced by the synthesis filter, as required by claim 1.

Furthermore, although Gao recognizes that the LPC coefficients *can be converted* to LSF (roots of the polynomials of the two decomposed transfer functions of the inverse filter transfer function A(z), one having even symmetry and the other having odd symmetry), nothing in Gao discloses *selecting roots* of the polynomial for one excitation function that minimizes a synthesis error produced by the synthesis filter. Therefore, Gao fails to disclose this limitation of the claim.

In sum, neither Kroon nor Gao, alone or in combination, disclose performing synthesis filter optimization, which includes including *selecting one of a plurality of excitation functions* **and** *selecting roots of the synthesis polynomial for one excitation function that minimizes a synthesis error produced by the synthesis filter,* as required by claim 1.

It is respectfully submitted that neither Kroon nor Gao, alone or in combination,

disclose each and every limitations of claim 1. Accordingly, Appellant respectfully requests that the rejection of claims 1, 3-7, and 12 under 35 U.S.C. §103(a) be withdrawn.

D. <u>Claim 2 is patentable over Kroon in view of Gao.</u>

Claim 2 depends directly from independent claim 1. The reasons cited above with respect to claim 1 are applicable to claim 2 and are herein incorporated by reference. Based on at least these reasons, claim 2 is patentable over Kroon in view of Gao.

In addition, for example, claim 2 requires optimizing roots of a synthesis filter polynomial using an iterative root optimization algorithm in response to the computed synthesized speech. This limitation is not disclosed by Kroon and Gao.

The Office action asserted that the limitation of claim 2 is taught by the polynomials shown in col. 13-14 of Kroon, and that line spectral frequencies may be used because they have known polynomial representations which are easily solved in such a way that the mathematical relationships between conjugate pairs may be exploited in order to solve them using known recursive ("iterative) techniques. See Office action, mailed January 27, 2006, page 6.

Although Kroon discloses that the LP coefficients are transformed to the LSP domain, they are done so only for quantization and interpolation purposes, and that the interpolated quantized and unquantized filters are converted back to the LP filter coefficients to construct the synthesis and weighting filters at each subframe. See col. 14, lines 12-18. See also col. 15, lines 22-26, which indicates that the LSP coefficients are defined as the roots of the sum and difference polynomials, and that the LP filter coefficients are converted to the line spectral pair (LSP) representation for quantization and interpolation purposes. So

although Kroon discloses that LP filter coefficients are converted to LSP coefficients (roots) for quantization and interpolation purposes, nothing in Kroon discloses optimizing roots of a synthesis filter polynomial using an iterative root optimization algorithm in response to the computed synthesized speech, as required by claim 2.

Gao, however, fails to cure the deficiency of Kroon. As described above, Gao does disclose a joint optimization; however, the joint optimization of Gao, as indicated by blocks 307 and 309 of Figure 3, identifies the optimum gain for the excitation signals output from the adaptive and fixed codebooks (ACB and FCB). Gao does in fact discloses that the LSP (which are also called Line Spectral Frequencies) is a transformation of LPC parameters, and that the LSPs are obtained by decomposing the inverse filter transfer function A(z) to a set of two transfer functions, one having even symmetry and the other having odd symmetry. The resulting LSPs are the roots of the two transfer functions. See col. 47, lines 57-64. However, Gao also discloses that the encoder analyzes each speech signal to extract the parameters of the CELP model, i.e., the *LP filter coefficients*, (LPC parameters) adaptive and fixed codebook indices and gains. These parameters are encoded and transmitted every subframe of the speech frame. In addition, LP analysis at the block 239 is performed twice per frame *but only a single set of LP parameters is converted to line spectrum frequencies (LSF) and vector quantized* using predictive multi-stage quantization (PMVQ). See col. 8, line 53 to col. 9, line 6. Gao, however, does not disclose that the roots or LSFs are used to perform synthesis filter optimization. The optimization of the gains of the ACB and the FCB specifically refer to using the LPC coefficients. See col. 6, lines 5-39, specifically lines 14-17. Accordingly, Gao only discloses that the synthesis filter optimization is performed using LPC coefficients, and not optimizing roots of a synthesis filter polynomial using an iterative

root optimization algorithm in response to the computed synthesized speech, as required by claim 2.

Furthermore, although Gao recognizes that the LPC coefficients *can be converted* to LSF (roots of the polynomials of the two decomposed transfer functions of the inverse filter transfer function A(z), one having even symmetry and the other having odd symmetry), nothing in Gao discloses *selecting roots* of the polynomial for one excitation function that minimizes a synthesis error produced by the synthesis filter. Therefore, Gao fails to disclose this limitation of the claim.

In sum, neither Kroon nor Gao, alone or in combination, disclose optimizing roots of a synthesis filter polynomial using an iterative root optimization algorithm in response to the computed synthesized speech, as required by claim 2.

It is respectfully submitted that neither Kroon nor Gao, alone or in combination, disclose each and every limitations of claim 2. Therefore, in addition to the reasons applied to their respective independent claim 1 are independently patentable over Kroon in view of Gao. Accordingly, Appellant respectfully requests that the rejection of claim 2 under 35 U.S.C. §103(a) be withdrawn.

E. <u>Claim 8 is patentable over Kroon in view of Gao.</u>

Claim 8 depends directly from independent claim 1. The reasons cited above with respect to claim 1 are applicable to claim 8 and are herein incorporated by reference. Based on at least these reasons, claim 8 is patentable over Kroon in view of Gao.

In addition, for example, claim 8 requires that excitation function is generated within an analysis frame comprising a plurality of speech samples; and where the synthesized

speech is computed in response to the samples which comprise at least one of the pulses and not in response to the samples which comprise none of the pulses. These limitations are not disclosed by Kroon and Gao.

The Office action contends that Kroon discloses this limitation in Figure 1 of Kroon, stating that "Figure 1 shows pulses more than one pulse which is therefore multipulses and spaces do not include pulses (unlike spaces in the prior art, fig. 2 that do not include a pulse)." See Office action, mailed January 27, 2006, pages 6-7. However, Appellant respectfully submits that nothing in Kroon discloses "excitation function is generated within an analysis frame comprising a plurality of speech samples."

As described above, Kroon discloses that an excitation signal is generated, however, this excitation signal does not include a number of non-zero pulses separated by spaces therebetween in an analysis frame. Kroon discloses that decoder generates an excitation signal based on a classification whether a speech signal is periodic (i.e., voiced) or non-periodic (i.e., unvoiced). See col. 2, lines 54-60. More specifically, when the speech signal is classified as periodic, the excitation signal is generated based on the output of a first portion (e.g., active codebook (ACB)) of the decoder, and not on the output signal of a second portion (e.g., active codebook (ACB)) of the decoder. Conversely, when the speech signal is classified as non-periodic, the excitation signal is generated based on the output of the second portion, and not on the output signal of the first portion. See col. 2, lines 61-67. Consequently, Kroon does not disclose that the excitation signal is generated within an analysis frame comprising a plurality of speech samples, and that the synthesized speech is computed in response to the samples which comprise at least one of the pulses and not in

response to the samples which comprise none of the pulses, but only that the excitation signal is either based on the output signal of either the first or second portions of the decoder.

Gao fails to cure the deficiency described above with respect to Kroon. Gao discloses that the excitation signal for an LPC synthesis filter is built from the two traditional components: 1) the pitch contribution; and 2) the innovation contribution. The pitch contribution is provided through use of an adaptive codebook. An innovation or fixed codebook has several subcodebooks in order to provide robustness against a wide range of input signals. To each of the two contributions a gain is applied which, multiplied with their respective codebook vectors and summed, to provide the excitation signal. See col. 40, lines 52-60. Consequently, Gao does not disclose that excitation signal is generated within an analysis frame comprising a plurality of speech samples, and that the synthesized speech is computed in response to the samples which comprise at least one of the pulses and not in response to the samples which comprise none of the pulses, but only that the excitation signal is the summation of the excitation vectors from the ACB and the FCB. Accordingly, Gao fails to cure the deficiency noted above with respect to Kroon.

Therefore, neither Kroon nor Gao, alone or in combination, disclose this limitation of the claim.

It is respectfully submitted that neither Kroon nor Gao, alone or in combination, disclose each and every limitations of claim 8. Therefore, in addition to the reasons applied to their respective independent claim 1 are independently patentable over Kroon in view of Gao. Accordingly, Appellant respectfully requests that the rejection of claim 8 under 35 U.S.C. §103(a) be withdrawn.

F. Claim 9-11 and 13-15 are patentable over Kroon in view of Gao, and further in view of Chen.

Claims 9-11 and 13-15 depend, directly or indirectly, from independent claim 1. The reasons cited above with respect to claim 1 are applicable to claims 9-11 and 13-15 and are herein incorporated by reference. Based on at least these reasons, claims 9-11 and 13-15 are patentable over Kroon in view of Gao.

In addition, the Office action contends that even though Kroon does not explicitly teach the particular claimed equations, Kroon teaches that it is well known to use polynomial (col. 13) root solutions in combination with multipulse (his one or more main pulses) Linear Prediction speech coding systems (co. 1, lines 55-64) and that Chen teaches that it is well known to optimize both the excitation and the LPC parameters using the identities on page 561, stating that Chen discloses "[re-optimizing] the synthesis filter parameters and pulse amplitudes in the Multi-Pulse Excitation Linear Prediction Synthesizer." See Office action, mailed January 27, 2006, page 7. However, Appellant respectfully submits that nothing in Kroon, Gao, or Chen disclose these limitations.

Although Chen disclose re-optimizing the synthesis filter parameters and pulse amplitudes in the Multi-Pulse Excitation Linear Prediction Synthesizer, Chen, however, discloses an optimizer that operates in a coefficient domain, not the root domain. See Chen, page 560-561. Operating in the root domain is much faster mathematically and more stable.

Furthermore, as described above with respect to claim 1, although Kroon discloses that the LP coefficients are transformed to the LSP domain, they are done so only for quantization and interpolation purposes, and that the interpolated quantized and unquantized filters are converted back to the LP filter coefficients to construct the synthesis and weighting

filters at each subframe. See col. 14, lines 12-18. See also col. 15, lines 22-26, which indicates that the LSP coefficients are defined as the roots of the sum and difference polynomials, and that the LP filter coefficients are converted to the line spectral pair (LSP) representation for quantization and interpolation purposes. So although Kroon discloses that LP filter coefficients are converted to LSP coefficients (roots) for quantization and interpolation purposes, nothing in Kroon discloses optimizing roots of a synthesis filter polynomial using an iterative root optimization algorithm in response to the computed synthesized speech, as required by claims 9-11 and 13-15.

Gao fails to cure the deficiency of Kroon. As described above, Gao does disclose a joint optimization; however, the joint optimization of Gao, as indicated by blocks 307 and 309 of Figure 3, identifies the optimum gain for the excitation signals output from the adaptive and fixed codebooks (ACB and FCB). Gao does in fact discloses that the LSP (which are also called Line Spectral Frequencies) is a transformation of LPC parameters, and that the LSPs are obtained by decomposing the inverse filter transfer function A(z) to a set of two transfer functions, one having even symmetry and the other having odd symmetry. The resulting LSPs are the roots of the two transfer functions. See col. 47, lines 57-64. However, Gao also discloses that the encoder analyzes each speech signal to extract the parameters of the CELP model, i.e., the *LP filter coefficients*, (LPC parameters) adaptive and fixed codebook indices and gains. These parameters are encoded and transmitted every subframe of the speech frame. In addition, LP analysis at the block 239 is performed twice per frame *but only a single set of LP parameters is converted to line spectrum frequencies (LSF) and vector quantized* using predictive multi-stage quantization (PMVQ). See col. 8, line 53 to col. 9, line 6. Gao, however, does not disclose that the roots or LSFs are used to perform

synthesis filter optimization. The optimization of the gains of the ACB and the FCB specifically refer to using the LPC coefficients.. See col. 6, lines 5-39, specifically lines 14-17. Accordingly, Gao only discloses that the synthesis filter optimization is performed using LPC coefficients, and not optimizing roots of a synthesis filter polynomial using an iterative root optimization algorithm in response to the computed synthesized speech, as required by claims 9-11 and 13-15.

Furthermore, although Gao recognizes that the LPC coefficients *can be converted* to LSF (roots of the polynomials of the two decomposed transfer functions of the inverse filter transfer function A(z), one having even symmetry and the other having odd symmetry), nothing in Gao discloses *selecting roots* of the polynomial for one excitation function that minimizes a synthesis error produced by the synthesis filter.

Since, both Gao and Kroon only disclose converting the LPC coefficients to the LSP coefficients (roots) for quantization and interpolation purposes, and Gao fails to cure the deficiencies of Gao and Kroon because it only disclose operation in the coefficient domain, the combination of Kroon, Gao, and Chen does not disclose each of the limitations in the claims.

It is respectfully submitted that neither Kroon, Gao, nor Chen, alone or in combination, disclose each and every limitations of claims 9-11 and 13-15. Therefore, in addition to the reasons applied to their respective independent claim 1 are independently patentable over Kroon in view of Gao, and further in view of Chen. Accordingly, Appellant respectfully requests that the rejections of claims 9-11 and 13-15 under 35 U.S.C. §103(a) be withdrawn.

G. Claims 16 is patentable over Kroon in view of Gao.

As described above, independent claim 16 includes limitations of producing a series of pulses within an analysis frame, where adjacent pulses define a space therebetween, and generating a synthesis polynomial, where generating the synthesis polynomial comprises calculating a contribution of the pulses and not calculating a contribution of only the space, and including selecting one of a plurality of excitation functions and selecting roots of the synthesis polynomial for the one excitation function that minimizes a synthesis error produced by the synthesis filter.

To establish *prima facie* obviousness of a claimed invention, all the claim limitations must be taught or suggested by the prior art. See M.P.E.P. §2143.03, citing *In re Royka*, 490 F.2d 981, 180, USPQ 580 (CCPA 1974). Appellants respectfully submit that neither Kroon nor Gao, alone or in combination, disclose all the limitations of claim 16.

First, Appellant respectfully submits that claim 16 requires the operation of "producing a series of pulses within an analysis frame, adjacent pulses defining a space therebetween." Neither Kroon nor Gao, alone or in combination, disclose this limitation of the claim.

The Office action contends that Kroon discloses this limitation in Figure 1 of Kroon, stating that "Figure 1 shows pulses upon which computations must be made in combination with spaces which do not require any computations and is used by Kroon as background showing how earlier systems (fig. 2) used a simpler pattern of excitation pulses which merely repeated. See Office action, mailed January 27, 2006, pages 5-6. However, Appellant

respectfully disagrees with the Office action that Kroon discloses "producing a series of pulses within an analysis frame, adjacent pulses defining a space therebetween."

Kroon is directed to a decoder for voiced/unvoiced classification of speech for excitation codebook selection in CELP speech decoding during frame erasures. See Kroon, Title and Abstract. Kroon discloses that an excitation signal is generated, however, this excitation signal does not include a number of non-zero pulses separated by spaces therebetween in an analysis frame. Kroon discloses that decoder generates an excitation signal based on a classification whether a speech signal is periodic (i.e., voiced) or non-periodic (i.e., unvoiced). See col. 2, lines 54-60. More specifically, when the speech signal is classified as periodic, the excitation signal is generated based on the output of a first portion (e.g., active codebook (ACB)) of the decoder, and not on the output signal of a second portion (e.g., active codebook (ACB)) of the decoder. Conversely, when the speech signal is classified as non-periodic, the excitation signal is generated based on the output of the second portion, and not on the output signal of the first portion. See col. 2, lines 61-67. Consequently, Kroon does not disclose producing a series of pulses within an analysis frame, adjacent pulses defining a space therebetween, but only that the excitation signal is either based on the output signal of either the first or second portions of the decoder.

Gao fails to cure the deficiency described above with respect to Kroon. Gao is directed to a speech encoder using warping in long term preprocessing. See Gao, Abstract. The encoder of Gao, to support lower bit rate encoding modes, may classify the input signal as either voiced or unvoiced speech, similarly to the decoder described in Kroon. See Figure 2 and col. 6, lines 5-39. In particular, the speech encoding circuitry simultaneously uses both the adaptive and fixed codebook (ACB and FCB) excitation vectors (i.e., signals) to

minimize a third error signal. See col. 6, lines 40-45. Gao also discloses that the excitation signal for an LPC synthesis filter is built from the two traditional components: 1) the pitch contribution; and 2) the innovation contribution. The pitch contribution is provided through use of an adaptive codebook. An innovation or fixed codebook has several subcodebooks in order to provide robustness against a wide range of input signals. To each of the two contributions a gain is applied which, multiplied with their respective codebook vectors and summed, provide the excitation signal. See col. 40, lines 52-60. Consequently, Gao does not disclose producing a series of pulses within an analysis frame, adjacent pulses defining a space therebetween, but only that the excitation signal is the summation of the excitation vectors from the ACB and the FCB. Accordingly, Gao fails to cure the deficiency noted above with respect to Kroon.

Therefore, neither Kroon nor Gao, alone or in combination, disclose this limitation of the claim.

Second, Appellant respectfully submits that claim 16 requires the operation of generating a synthesis polynomial, where generating the synthesis polynomial comprises calculating a contribution of the pulses and not calculating a contribution of only the space, and including selecting one of a plurality of excitation functions and selecting roots of the synthesis polynomial for the one excitation function that minimizes a synthesis error produced by the synthesis filter." Neither Kroon nor Gao, alone or in combination, disclose this limitation of the claim.

The Office action asserted that computing a synthesis polynomial is shown by Kroon in col. 13, line 37 to col. 14, where he explicitly teaches using the well know mathematical relationships with polynomials and root solutions techniques in a Linear Prediction

synthesizer. Appellant respectfully disagrees with the Office action that Kroon discloses this limitation of the claim.

Although Kroon discloses that the LP coefficients are transformed to the LSP domain, they are done so only for quantization and interpolation purposes, and that the interpolated quantized and unquantized filters are converted back to the LP filter coefficients to construct the synthesis and weighting filters at each subframe. See col. 14, lines 12-18. See also col. 15, lines 22-26, which indicates that the LSP coefficients are defined as the roots of the sum and difference polynomials, and that the LP filter coefficients are converted to the line spectral pair (LSP) representation for quantization and interpolation purposes. So although Kroon discloses that LP filter coefficients are converted to LSP coefficients (roots) for quantization and interpolation purposes, nothing in Kroon discloses that the LSP coefficients are roots of the synthesis polynomial and that they are selected for the one excitation function that minimizes a synthesis error produced by the synthesis filter, as required by claim 16.

Gao, however, fails to cure the deficiency of Kroon. As described above, Gao does disclose a joint optimization; however, the joint optimization of Gao, as indicated by blocks 307 and 309 of Figure 3, identifies the optimum gain for the excitation signals output from the adaptive and fixed codebooks (ACB and FCB). This is done by generating a synthesized and weighted signal via block 301 and 303, that best matches the first target signal 229, which minimizes the third error signal. In other words, Gao only discloses joint optimization of the gains for both the adaptive and fixed codebooks vector selections to minimize the error signal.

The Office action also contends that this limitation is met because Gao discloses that the use of Line Spectral Pair (LSP) inherently performs transformation of LPC parameters which are roots of these polynomials on the z-unit circle. See Office action, mailed January 27, page 6. Gao does in fact discloses that the LSP (which are also called Line Spectral Frequencies) is a transformation of LPC parameters, and that the LSPs are obtained by decomposing the inverse filter transfer function A(z) to a set of two transfer functions, one having even symmetry and the other having odd symmetry. The resulting LSPs are the roots of the two transfer functions. See col. 47, lines 57-64. However, Gao also discloses that the encoder analyzes each speech signal to extract the parameters of the CELP model, i.e., the *LP filter coefficients*, (LPC parameters) adaptive and fixed codebook indices and gains. These parameters are encoded and transmitted every subframe of the speech frame. In addition, LP analysis at the block 239 is performed twice per frame *but only a single set of LP parameters is converted to line spectrum frequencies (LSF) and vector quantized* using predictive multi-stage quantization (PMVQ). See col. 8, line 53 to col. 9, line 6. Gao, however, does not disclose that the roots or LSFs are used to perform synthesis filter optimization. The optimization of the gains of the ACB and the FCB specifically refer to using the LPC coefficients. See col. 6, lines 5-39, specifically lines 14-17. Accordingly, Gao only discloses that the synthesis filter optimization is performed using LPC coefficients, and not by selecting roots of the synthesis polynomial for one excitation function that minimizes a synthesis error produced by the synthesis filter, as required by claim 16.

Furthermore, although Gao recognizes that the LPC coefficients *can be converted* to LSF (roots of the polynomials of the two decomposed transfer functions of the inverse filter transfer function A(z), one having even symmetry and the other having odd symmetry),

nothing in Gao discloses *selecting roots* of the polynomial for one excitation function that minimizes a synthesis error produced by the synthesis filter. Therefore, Gao fails to disclose this limitation of the claim.

In sum, neither Kroon nor Gao, alone or in combination, disclose generating a synthesis polynomial, where generating the synthesis polynomial comprises calculating a contribution of the pulses and not calculating a contribution of only the space, and including selecting one of a plurality of excitation functions and *selecting roots of the synthesis polynomial for one excitation function that minimizes a synthesis error produced by the synthesis filter,* as required by claim 16.

H. Claims 17 is patentable over Kroon in view of Gao.

Claim 17 depends directly from independent claim 16. The reasons cited above with respect to claim 16 are applicable to claim 17 and are herein incorporated by reference. Based on at least these reasons, claim 17 is patentable over Kroon in view of Gao.

In addition, for example, claim 17 requires that the synthesis polynomial computation includes calculating a convolution of an impulse response and the excitation function; wherein the excitation function is generated within an analysis frame comprising a plurality of speech samples; and wherein the synthesis polynomial is computed in response to the samples which comprise at least one of the pulses and is not computed in response to the samples which comprise none of the pulses; and further comprising optimizing roots of the synthesis polynomial using an iterative root optimization algorithm. Neither Kroon nor Gao, alone or in combination, disclose this limitation of the claim.

The Office action asserted that computing a synthesis polynomial is shown by Kroon

in col. 13, lines 37-col. 14, stating that Kroon "teaches using the well known mathematical relationships with polynomials and root solutions techniques in a Linear Prediction synthesizer. See Office action, mailed January 27, 2006, page 7. However, Appellant respectfully submits that nothing in Kroon discloses "that the synthesis polynomial computation includes calculating a convolution of an impulse response and the excitation function; wherein the excitation function is generated within an analysis frame comprising a plurality of speech samples; and wherein the synthesis polynomial is computed in response to the samples which comprise at least one of the pulses and is not computed in response to the samples which comprise none of the pulses; and further comprising optimizing roots of the synthesis polynomial using an iterative root optimization algorithm."

First, as described above with respect to claim 8, Kroon discloses that an excitation signal is generated, however, this excitation signal does not include a number of non-zero pulses separated by spaces therebetween in an analysis frame. Kroon discloses that decoder generates an excitation signal based on a classification whether a speech signal is periodic (i.e., voiced) or non-periodic (i.e., unvoiced). See col. 2, lines 54-60. More specifically, when the speech signal is classified as periodic, the excitation signal is generated based on the output of a first portion (e.g., active codebook (ACB)) of the decoder, and not on the output signal of a second portion (e.g., active codebook (ACB)) of the decoder. Conversely, when the speech signal is classified as non-periodic, the excitation signal is generated based on the output of the second portion, and not on the output signal of the first portion. See col. 2, lines 61-67. Consequently, Kroon does not disclose that the excitation signal is generated within an analysis frame comprising a plurality of speech samples, and that the synthesized speech is computed in response to the samples which comprise at least one of the pulses and

not in response to the samples which comprise none of the pulses, but only that the excitation signal is either based on the output signal of either the first or second portions of the decoder.

Gao fails to cure the deficiency described above with respect to Kroon. Gao discloses that the excitation signal for an LPC synthesis filter is built from the two traditional components: 1) the pitch contribution; and 2) the innovation contribution. The pitch contribution is provided through use of an adaptive codebook. An innovation or fixed codebook has several subcodebooks in order to provide robustness against a wide range of input signals. To each of the two contributions a gain is applied which, multiplied with their respective codebook vectors and summed, to provide the excitation signal. See col. 40, lines 52-60. Consequently, Gao does not disclose that excitation signal is generated within an analysis frame comprising a plurality of speech samples, and that the synthesized speech is computed in response to the samples which comprise at least one of the pulses and not in response to the samples which comprise none of the pulses, but only that the excitation signal is the summation of the excitation vectors from the ACB and the FCB. Accordingly, Gao fails to cure the deficiency noted above with respect to Kroon.

Therefore, neither Kroon nor Gao, alone or in combination, disclose this limitation of the claim.

Second, as described above with respect to claim 2, although Kroon discloses that the LP coefficients are transformed to the LSP domain, they are done so only for quantization and interpolation purposes, and that the interpolated quantized and unquantized filters are converted back to the LP filter coefficients to construct the synthesis and weighting filters at each subframe. See col. 14, lines 12-18. See also col. 15, lines 22-26, which indicates that the LSP coefficients are defined as the roots of the sum and difference polynomials, and that

the LP filter coefficients are converted to the line spectral pair (LSP) representation for quantization and interpolation purposes. So although Kroon discloses that LP filter coefficients are converted to LSP coefficients (roots) for quantization and interpolation purposes, nothing in Kroon discloses optimizing roots of a synthesis filter polynomial using an iterative root optimization algorithm in response to the computed synthesized speech, as required by claim 17.

Gao, however, fails to cure the deficiency of Kroon. As described above, Gao does disclose a joint optimization; however, the joint optimization of Gao, as indicated by blocks 307 and 309 of Figure 3, identifies the optimum gain for the excitation signals output from the adaptive and fixed codebooks (ACB and FCB). Gao does in fact discloses that the LSP (which are also called Line Spectral Frequencies) is a transformation of LPC parameters, and that the LSPs are obtained by decomposing the inverse filter transfer function A(z) to a set of two transfer functions, one having even symmetry and the other having odd symmetry. The resulting LSPs are the roots of the two transfer functions. See col. 47, lines 57-64. However, Gao also discloses that the encoder analyzes each speech signal to extract the parameters of the CELP model, i.e., the *LP filter coefficients*, (LPC parameters) adaptive and fixed codebook indices and gains. These parameters are encoded and transmitted every subframe of the speech frame. In addition, LP analysis at the block 239 is performed twice per frame *but only a single set of LP parameters is converted to line spectrum frequencies (LSF) and vector quantized* using predictive multi-stage quantization (PMVQ). See col. 8, line 53 to col. 9, line 6. Gao, however, does not disclose that the roots or LSFs are used to perform synthesis filter optimization. The optimization of the gains of the ACB and the FCB specifically refer to using the LPC coefficients. See col. 6, lines 5-39, specifically lines 14-

17. Accordingly, Gao only discloses that the synthesis filter optimization is performed using LPC coefficients, and not optimizing roots of a synthesis filter polynomial using an iterative root optimization algorithm in response to the computed synthesized speech, as required by claim 17.

Furthermore, although Gao recognizes that the LPC coefficients *can be converted* to LSF (roots of the polynomials of the two decomposed transfer functions of the inverse filter transfer function A(z), one having even symmetry and the other having odd symmetry), nothing in Gao discloses *selecting roots* of the polynomial for one excitation function that minimizes a synthesis error produced by the synthesis filter. Therefore, Gao fails to disclose this limitation of the claim.

Third, Kroon discloses a convolution operation, however, the convolution operation of Kroon does not perform the convolution of an impulse response and the excitation function, but rather convolves the past excitation at delay k with the impulse response. See Kroon, col. 20, lines 22 to col. 22, line 26, specifically col. 21, lines 29-55. Therefore, Kroon fails to disclose this limitation of the claim, and Gao fails to cure this deficiency.

In sum, neither Kroon nor Gao, alone or in combination, disclose that the synthesis polynomial computation includes calculating a convolution of an impulse response and the excitation function; wherein the excitation function is generated within an analysis frame comprising a plurality of speech samples; and wherein the synthesis polynomial is computed in response to the samples which comprise at least one of the pulses and is not computed in response to the samples which comprise none of the pulses; and further comprising optimizing roots of the synthesis polynomial using an iterative root optimization algorithm, as required by claim 17.

It is respectfully submitted that neither Kroon nor Gao, alone or in combination, disclose each and every limitations of claim 17. Therefore, in addition to the reasons applied to their respective independent claim 16 are independently patentable over Kroon in view of Gao. Accordingly, Appellant respectfully requests that the rejection of claim 17 under 35 U.S.C. §103(a) be withdrawn.

I. <u>Claim 18 is patentable over Kroon in view of Gao, and further in view of Chen.</u>

Claim 18 depends directly from independent claim 16. The reasons cited above with respect to claim 16 are applicable to claim 18 and are herein incorporated by reference. Based on at least these reasons, claim 18 is patentable over Kroon in view of Gao.

In addition, the Office action contends that even though Kroon does not explicitly teach the particular claimed equations, Kroon teaches that it is well known to use polynomial (col. 13) root solutions in combination with multipulse (his one or more main pulses) Linear Prediction speech coding systems (co. 1, lines 55-64) and that Chen teaches that it is well known to optimize both the excitation and the LPC parameters using the identities on page 561, stating that Chen discloses "[re-optimizing] the synthesis filter parameters and pulse amplitudes in the Multi-Pulse Excitation Linear Prediction Synthesizer." See Office action, mailed January 27, 2006, page 7. However, Appellant respectfully submits that nothing in Kroon, Gao, or Chen disclose these limitations.

Although Chen disclose re-optimizing the synthesis filter parameters and pulse amplitudes in the Multi-Pulse Excitation Linear Prediction Synthesizer, Chen, however, discloses an optimizer that operates in a coefficient domain, not the root domain. See Chen, page 560-561. Operating in the root domain is much faster mathematically and more stable.

Furthermore, as described above with respect to claim 1, although Kroon discloses that the LP coefficients are transformed to the LSP domain, they are done so only for quantization and interpolation purposes, and that the interpolated quantized and unquantized filters are converted back to the LP filter coefficients to construct the synthesis and weighting filters at each subframe. See col. 14, lines 12-18. See also col. 15, lines 22-26, which indicates that the LSP coefficients are defined as the roots of the sum and difference polynomials, and that the LP filter coefficients are converted to the line spectral pair (LSP) representation for quantization and interpolation purposes. So although Kroon discloses that LP filter coefficients are converted to LSP coefficients (roots) for quantization and interpolation purposes, nothing in Kroon discloses optimizing roots of a synthesis filter polynomial using an iterative root optimization algorithm in response to the computed synthesized speech, as required by claim 18.

Gao fails to cure the deficiency of Kroon. As described above, Gao does disclose a joint optimization; however, the joint optimization of Gao, as indicated by blocks 307 and 309 of Figure 3, identifies the optimum gain for the excitation signals output from the adaptive and fixed codebooks (ACB and FCB). Gao does in fact discloses that the LSP (which are also called Line Spectral Frequencies) is a transformation of LPC parameters, and that the LSPs are obtained by decomposing the inverse filter transfer function A(z) to a set of two transfer functions, one having even symmetry and the other having odd symmetry. The resulting LSPs are the roots of the two transfer functions. See col. 47, lines 57-64. However, Gao also discloses that the encoder analyzes each speech signal to extract the parameters of the CELP model, i.e., the *LP filter coefficients*, (LPC parameters) adaptive and fixed codebook indices and gains. These parameters are encoded and transmitted every subframe

of the speech frame. In addition, LP analysis at the block 239 is performed twice per frame *but only a single set of LP parameters is converted to line spectrum frequencies (LSF) and vector quantized* using predictive multi-stage quantization (PMVQ). See col. 8, line 53 to col. 9, line 6. Gao, however, does not disclose that the roots or LSFs are used to perform synthesis filter optimization. The optimization of the gains of the ACB and the FCB specifically refer to using the LPC coefficients. See col. 6, lines 5-39, specifically lines 14-17. Accordingly, Gao only discloses that the synthesis filter optimization is performed using LPC coefficients, and not optimizing roots of a synthesis filter polynomial using an iterative root optimization algorithm in response to the computed synthesized speech, as required by claim 18.

Furthermore, although Gao recognizes that the LPC coefficients *can be converted* to LSF (roots of the polynomials of the two decomposed transfer functions of the inverse filter transfer function A(z), one having even symmetry and the other having odd symmetry), nothing in Gao discloses *selecting roots* of the polynomial for one excitation function that minimizes a synthesis error produced by the synthesis filter.

Since, both Gao and Kroon only disclose converting the LPC coefficients to the LSP coefficients (roots) for quantization and interpolation purposes, and Gao fails to cure the deficiencies of Gao and Kroon because it only disclose operation in the coefficient domain, the combination of Kroon, Gao, and Chen does not disclose each of the limitations in the claims.

It is respectfully submitted that neither Kroon, Gao, nor Chen, alone or in combination, disclose each and every limitations of claim 18. Therefore, in addition to the reasons applied to their respective independent claim 1 are independently patentable over

Kroon in view of Gao, and further in view of Chen. Accordingly, Appellant respectfully requests that the rejection of claim 18 under 35 U.S.C. §103(a) be withdrawn.

J. Claim 19 is patentable over Kroon in view of Gao.

As described above, independent claim 19 includes limitations of an excitation module responsive to an original speech and generating an excitation function using an excitation module, where the excitation function comprises a series of pulses within an analysis frame; and a synthesis filter responsive to the excitation function and the original speech and generating a synthesized speech using the synthesis filter; wherein the synthesis filter computes a convolution of an impulse response and the excitation function, the convolution computation comprises calculating samples of speech having only the pulses within the analysis frame; including selecting one of a plurality of excitation functions and selecting roots of the synthesis polynomial for the one excitation function that minimizes a synthesis error produced by the synthesis filter.

To establish *prima facie* obviousness of a claimed invention, all the claim limitations must be taught or suggested by the prior art. See M.P.E.P. §2143.03, citing *In re Royka*, 490 F.2d 981, 180, USPQ 580 (CCPA 1974). Appellants respectfully submit that neither Kroon nor Gao, alone or in combination, disclose all the limitations of claim 19.

Appellant respectfully submits that claim 19 requires that the synthesis filter computes a convolution of an impulse response and the excitation function, the convolution computation comprises calculating samples of speech having only the pulses within the analysis frame; and that the synthesis filter selects one of a plurality of excitation functions

and roots of the synthesis polynomial for the one excitation function that minimizes a synthesis error produced by the synthesis filter. Neither Kroon nor Gao, alone or in combination, disclose these limitation of the claim.

The Office action asserted that computing a synthesis polynomial is shown by Kroon in col. 13, lines 37-col. 14, stating that Kroon "teaches using the well known mathematical relationships with polynomials and root solutions techniques in a Linear Prediction synthesizer. See Office action, mailed January 27, 2006, page 7. However, Appellant respectfully submits that nothing in Kroon discloses that the synthesis filter computes a convolution of an impulse response and the excitation function, where the convolution computation comprises calculating samples of speech having only the pulses within the analysis frame; and that the synthesis filter selects one of a plurality of excitation functions and roots of the synthesis polynomial for the one excitation function that minimizes a synthesis error produced by the synthesis filter.

First, as described with respect to claim 17, Kroon discloses a convolution operation, however, the convolution operation of Kroon does not perform calculating samples of speech having only said pulses within the analysis frame, but rather convolves the past excitation at delay k with the impulse response. See Kroon, col. 20, lines 22 to col. 22, line 26, specifically col. 21, lines 29-55. Therefore, Kroon fails to disclose this limitation of the claim, and Gao fails to cure this deficiency.

Second, as described with respect to claim 1, the Office action concedes that Kroon fails to disclose this limitation of the claim, and that Gao fails to cure this deficiency.

Gao does disclose a joint optimization; however, the joint optimization of Gao, as indicated by blocks 307 and 309 of Figure 3, identifies the optimum gain for the excitation

signals output from the adaptive and fixed codebooks (ACB and FCB). This is done by generating a synthesized and weighted signal via block 301 and 303, that best matches the first target signal 229, which minimizes the third error signal. In other words, Gao only discloses joint optimization of the gains for both the adaptive and fixed codebooks vector selections to minimize the error signal.

The Office action also asserted in the analysis of claim 1 that this limitation is met because Gao discloses that the use of Line Spectral Pair (LSP) inherently performs transformation of LPC parameters which are roots of these polynomials on the z-unit circle. See Office action, mailed January 27, page 6. Gao does in fact discloses that the LSP (which are also called Line Spectral Frequencies) is a transformation of LPC parameters, and that the LSPs are obtained by decomposing the inverse filter transfer function A(z) to a set of two transfer functions, one having even symmetry and the other having odd symmetry. The resulting LSPs are the roots of the two transfer functions. See col. 47, lines 57-64. However, Gao also discloses that the encoder analyzes each speech signal to extract the parameters of the CELP model, i.e., the *LP filter coefficients*, (LPC parameters) adaptive and fixed codebook indices and gains. These parameters are encoded and transmitted every subframe of the speech frame. In addition, LP analysis at the block 239 is performed twice per frame *but only a single set of LP parameters is converted to line spectrum frequencies (LSF) and vector quantized* using predictive multi-stage quantization (PMVQ). See col. 8, line 53 to col. 9, line 6. Gao, however, does not disclose that the roots or LSFs are used to perform synthesis filter optimization. The optimization of the gains of the ACB and the FCB specifically refer to using the LPC coefficients. See col. 6, lines 5-39, specifically lines 14-17. Accordingly, Gao only discloses that the synthesis filter optimization is performed using

LPC coefficients, and not by selecting roots of the synthesis polynomial for one excitation function that minimizes a synthesis error produced by the synthesis filter, as required by claim 19.

Furthermore, although Gao recognizes that the LPC coefficients *can be converted* to LSF (roots of the polynomials of the two decomposed transfer functions of the inverse filter transfer function A(z), one having even symmetry and the other having odd symmetry), nothing in Gao discloses *selecting roots* of the polynomial for one excitation function that minimizes a synthesis error produced by the synthesis filter. Therefore, Gao fails to disclose this limitation of the claim.

In sum, neither Kroon nor Gao, alone or in combination, disclose a synthesizer filter that computes a convolution of an impulse response and the excitation function, where the convolution computation comprises calculating samples of speech having only the pulses within the analysis frame; and that the synthesis filter selects one of a plurality of excitation functions and roots of the synthesis polynomial for the one excitation function that minimizes a synthesis error produced by the synthesis filter, as required by claim 19.

K. <u>Claims 20-25 are patentable over Kroon in view of Gao, and further in view of Chen.</u>

Claims 20-25 depend, directly or indirectly, from independent claim 19. The reasons cited above with respect to claim 19 are applicable to claims 20-25 and are herein incorporated by reference. Based on at least these reasons, claims 20-25 are patentable over Kroon in view of Gao.

In addition, the Office action contends that even though Kroon does not explicitly teach the particular claimed equations, Kroon teaches that it is well known to use polynomial (col. 13) root solutions in combination with multipulse (his one or more main pulses) Linear Prediction speech coding systems (co. 1, lines 55-64) and that Chen teaches that it is well known to optimize both the excitation and the LPC parameters using the identities on page 561, stating that Chen discloses "[re-optimizing] the synthesis filter parameters and pulse amplitudes in the Multi-Pulse Excitation Linear Prediction Synthesizer." See Office action, mailed January 27, 2006, page 7. However, Appellant respectfully submits that nothing in Kroon, Gao, or Chen disclose these limitations.

Although Chen disclose re-optimizing the synthesis filter parameters and pulse amplitudes in the Multi-Pulse Excitation Linear Prediction Synthesizer, Chen, however, discloses an optimizer that operates in a coefficient domain, not the root domain. See Chen, page 560-561. Operating in the root domain is much faster mathematically and more stable.

Furthermore, as described above with respect to claim 1, although Kroon discloses that the LP coefficients are transformed to the LSP domain, they are done so only for quantization and interpolation purposes, and that the interpolated quantized and unquantized filters are converted back to the LP filter coefficients to construct the synthesis and weighting filters at each subframe. See col. 14, lines 12-18. See also col. 15, lines 22-26, which indicates that the LSP coefficients are defined as the roots of the sum and difference polynomials, and that the LP filter coefficients are converted to the line spectral pair (LSP) representation for quantization and interpolation purposes. So although Kroon discloses that LP filter coefficients are converted to LSP coefficients (roots) for quantization and interpolation purposes, nothing in Kroon discloses optimizing roots of a synthesis filter

polynomial using an iterative root optimization algorithm in response to the computed synthesized speech, as required by claims 20-25.

Gao fails to cure the deficiency of Kroon. As described above, Gao does disclose a joint optimization; however, the joint optimization of Gao, as indicated by blocks 307 and 309 of Figure 3, identifies the optimum gain for the excitation signals output from the adaptive and fixed codebooks (ACB and FCB). Gao does in fact discloses that the LSP (which are also called Line Spectral Frequencies) is a transformation of LPC parameters, and that the LSPs are obtained by decomposing the inverse filter transfer function A(z) to a set of two transfer functions, one having even symmetry and the other having odd symmetry. The resulting LSPs are the roots of the two transfer functions. See col. 47, lines 57-64. However, Gao also discloses that the encoder analyzes each speech signal to extract the parameters of the CELP model, i.e., the *LP filter coefficients*, (LPC parameters) adaptive and fixed codebook indices and gains. These parameters are encoded and transmitted every subframe of the speech frame. In addition, LP analysis at the block 239 is performed twice per frame *but only a single set of LP parameters is converted to line spectrum frequencies (LSF) and vector quantized* using predictive multi-stage quantization (PMVQ). See col. 8, line 53 to col. 9, line 6. Gao, however, does not disclose that the roots or LSFs are used to perform synthesis filter optimization. The optimization of the gains of the ACB and the FCB specifically refer to using the LPC coefficients. See col. 6, lines 5-39, specifically lines 14-17. Accordingly, Gao only discloses that the synthesis filter optimization is performed using LPC coefficients, and not optimizing roots of a synthesis filter polynomial using an iterative root optimization algorithm in response to the computed synthesized speech, as required by claims 20-25.

Furthermore, although Gao recognizes that the LPC coefficients *can be converted* to LSF (roots of the polynomials of the two decomposed transfer functions of the inverse filter transfer function A(z), one having even symmetry and the other having odd symmetry), nothing in Gao discloses *selecting roots* of the polynomial for one excitation function that minimizes a synthesis error produced by the synthesis filter.

Since, both Gao and Kroon only disclose converting the LPC coefficients to the LSP coefficients (roots) for quantization and interpolation purposes, and Gao fails to cure the deficiencies of Gao and Kroon because it only disclose operation in the coefficient domain, the combination of Kroon, Gao, and Chen does not disclose each of the limitations in the claims.

It is respectfully submitted that neither Kroon, Gao, nor Chen, alone or in combination, disclose each and every limitations of claims 20-25. Therefore, in addition to the reasons applied to their respective independent claim 1 are independently patentable over Kroon in view of Gao, and further in view of Chen. Accordingly, Appellant respectfully requests that the rejections of claims 20-25 under 35 U.S.C. §103(a) be withdrawn.

VIII. CONCLUSION

For the reasons stated above, claims 1-25 are patentable. Appellant respectfully requests that the Board reverse the rejections of the claims 1-25and direct the Examiner to enter a Notice of Allowance for claims 1-25.

Enclosed is a check in the amount of $500.00 to cover the fee for filing a brief in support of an appeal as required under 37 C.F.R. § 1.17(c) and 41.20(b)(2).

Authorization is hereby given to charge our Deposit Account No. 02-2666 for any charges that may be due. Furthermore, if an extension is required, then Appellant hereby requests such extension.

Respectfully submitted,
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Dated: October 2, 2006

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APPENDIX A: Claims on Appeal

(37 C.F.R. § 41.37(c)(1)(viii))

The claims on appeal read as follows:

1. (Previously Presented) A method of digitally encoding speech, comprising

generating an excitation function using an excitation module, said excitation function comprising a number of non-zero pulses within an analysis frame separated by spaces therebetween;

generating synthesized speech using a synthesis filter from said number of non-zero pulses within the analysis frame without contribution from the spaces therebetween; and

performing synthesis filter optimization, including selecting one of a plurality of excitation functions and selecting roots of the synthesis polynomial for one excitation function that minimizes a synthesis error produced by the synthesis filter.

2. (Original) The method according to claim 1, further comprising optimizing roots of a synthesis filter polynomial using an iterative root optimization algorithm in response to said computed synthesized speech.

3. (Original) The method according to claim 1, wherein said pulses are non-uniformly spaced.

4. (Original) The method according to claim 1, wherein said pulses are uniformly spaced.

5. (Original) The method according to claim 1, wherein said excitation function is generated using a linear prediction coding ("LPC") encoder.

6. (Original) The method according to claim 1, wherein said excitation function is generated using a multipulse encoder.

7. (Original) The method according to claim 1, wherein said spaces comprise no pulses.

8. (Original) The method according to claim 1, wherein said excitation function is generated within an analysis frame comprising a plurality of speech samples; and wherein said synthesized speech is computed in response to said samples which comprise at least one of said pulses and not in response to said samples which comprise none of said pulses.

9. (Previously Presented) The method according to claim 1, wherein said synthesized speech is calculated using the formula:

$$\hat{s}(n) = h(n) * u(n) = \sum_{k=1}^{F(n)} h(n - p(k))u(p(k))$$

wherein $\hat{s}(n)$ is the synthesized speech sample at time n, h(n) is the impulse response of the synthesis filter at time n, u(n) is the excitation function at time n, and p(k) is a location of the k-the excitation pulse in the frame.

10. (Previously Presented) The method according to claim 9, wherein said synthesized speech is further calculated using the formula:

$$\hat{s}(n) = \sum_{k=0}^{n} h(k)u(n-k) = \sum_{k-1}^{F(n)} u(p(k)) \sum_{i=1}^{M} b_i(\lambda_i)^{n-p(k)}$$

where b_i is the i-th decomposition coefficient; and

where said excitation function is defined by the formulas:

$$u(p(k)) \neq 0 \quad \text{for} \quad k = 1,2 \ldots N_p$$

$$u(n) = 0 \qquad \text{for} \quad n \neq p(k)$$

and where F(n) is a number of excitation pulses in an analysis frame up to sample n

and

is defined by the formulas:

$$p(F(n)) \leq n$$

$F(n)) \leq N_p$, where N_p is the number of excitation pulses in the analysis frame.

11. (Previously Presented) The method according to claim 10, further comprising computing roots of a synthesis polynomial using the formula:

$$\partial\hat{s}(k)/\partial\lambda_r^{(j)} = b_r \sum_{m=1}^{F(k)} (k-p(m))u(p(m)) \, (\lambda_r^{(j)})^{(k-p(m)-1)}$$

where λ_r is the r-th root of the synthesis filter, $\lambda_r^{(j)}$ is the r-th root at the j-th iteration, and $\partial\hat{s}(k)/\partial\lambda_r^{(j)}$ is the partial derivative of the k-th synthesized speech sample relative to the r-th root of the synthesis filter at the j-th iteration.

12. (Original) The method according to claim 1, wherein said synthesized speech computation comprises calculating a convolution of an impulse response and said excitation function; and wherein said spaces comprise no pulses.

13. (Previously Presented) The method according to claim 12, wherein said excitation function is generated within an analysis frame comprising a plurality of speech samples; wherein said synthesized speech is computed in response to said samples which comprise at least one of said pulses and is not computed in response to said samples which comprise none of said pulses; and wherein said synthesized speech is calculated using the formula:

$$\hat{s}(n) = h(n) * u(n) = \sum_{k=1}^{F(n)} h(n - p(k))u(p(k))$$

wherein $\hat{s}(n)$ is the synthesized speech sample at time n, h(n) is the impulse response of the synthesis filter at time n, u(n) is the excitation function at time n, and p(k) is a location of the k-th excitation pulse in the frame.

14. (Original) The method according to claim 13, wherein said pulses are non-uniformly spaced; and wherein said excitation function is generated using a multipulse encoder.

15. (Original) The method according to claim 14, further comprising optimizing roots of a synthesis polynomial using an iterative root searching algorithm in response to said computed synthesized speech.

16. (Previously Presented) A method of digitally encoding speech, comprising

producing a series of pulses within an analysis frame, adjacent pulses defining a space therebetween; and

generating a synthesis polynomial, said generating the synthesis polynomial comprising calculating a contribution of said pulses and not calculating a contribution of only said space, and including selecting one of a plurality of excitation functions and selecting roots of the synthesis polynomial for the one excitation function that minimizes a synthesis error produced by the synthesis filter.

17. (Previously Presented) The method according to claim 16, wherein said synthesis polynomial computation comprises calculating a convolution of an impulse response and said excitation function; wherein said excitation function is generated within an analysis frame comprising a plurality of speech samples; and wherein said synthesis polynomial is computed in response to said samples which comprise at least one of said pulses and is not computed in response to said samples which comprise none of said pulses; and further comprising optimizing roots of said synthesis polynomial using an iterative root optimization algorithm.

18. (Previously Presented) The method according to claim 17, wherein said synthesis polynomial is calculated using the formula:

$$\hat{s}(n) = h(n) * u(n) = \sum_{k-1}^{F(n)} h(n - p(k))u(p(k))$$

wherein $\hat{s}(n)$ is the synthesized speech sample at time n, h(n) is the impulse response of the synthesis filter at time n, u(n) is the excitation function at time n, and p(k) is a location of the k-th excitation pulse in the frame; and

where said excitation function is defined by the formulas:

$$u(p(k)) \neq 0 \quad \text{for} \quad k = 1,2 \ldots N_p$$

$$u(n) = 0 \qquad \text{for} \quad n \neq p(k)$$

and where F(n) is a number of excitation pulses in an analysis frame up to sample n and is defined by the formulas:

$$p(F(n)) \leq n$$

$F(n)) \leq N_p$, where N_p is the number of excitation pulses in an analysis frame.

19. (Previously Presented) A speech synthesis system, comprising

an excitation module responsive to an original speech and generating an excitation function using an excitation module, said excitation function comprising a series of pulses within an analysis frame; and

a synthesis filter responsive to said excitation function and said original speech and generating a synthesized speech using a synthesis filter; wherein said synthesis filter computes a convolution of an impulse response and said excitation function, said convolution computation comprising calculating samples of speech having only said pulses within the analysis frame; including selecting one of a plurality of excitation functions and selecting roots of the synthesis polynomial for the one excitation function that minimizes a synthesis error produced by the synthesis filter.

20. (Previously Presented) The method according to claim 19, wherein said synthesis filter computes roots of a synthesis polynomial using the formula:

$$\partial\hat{s}(k)/\partial\lambda_r^{(j)} = b_r \sum_{m=1}^{F(k)} (k-p(m))u(p(m))\,(\lambda_r^{(j)})^{(k-p(m)-1)}$$

where λ_r is the r-th root of the synthesis filter, $\lambda_r^{(j)}$ is the r-th root at the j-th iteration, and $\partial\hat{s}(k)/\partial\lambda_r^{(j)}$ is the partial derivative of the k-th synthesized speech sample relative to the r-th synthesis filter at the j-th iteration, where p(m) is a location of the m-th excitation pulse, u(p(m)) is an excitation function at time p(m), and k is a time index.

21. (Previously Presented) The method according to claim 19, wherein said convolution computation is calculated using the formula:

$$\hat{s}(n) = \sum_{k=0}^{n} h(k)u(n-k) = \sum_{k-1}^{F(n)} u(p(k)) \sum_{i=1}^{M} b_i(\lambda_i)^{n-p(k)}$$

where λ_r is the r-th root of the synthesis filter, $\lambda_r^{(j)}$ is the r-th root at the j-th iteration, and $\partial\hat{s}(k)/\partial\lambda_r^{(j)}$ is the partial derivative of the k-th synthesized speech sample relative to the r-th synthesis filter at the j-th iteration, where p(m) is a location of the m-th excitation pulse, u(p(k)) is an excitation function at time p(k), and k is a time index; and

where said excitation function is defined by the formulas:

$$u(p(k)) \neq 0 \quad \text{for} \quad k = 1,2 \ldots N_p$$

$$u(n) = 0 \qquad \text{for} \quad n \neq p(k)$$

and where F(n) is a number of excitation pulses in an analysis frame up to sample n and is defined by the formulas:

$p(F(n)) \leq n$

$F(n)) \leq N_p$, where N_p is the number of excitation pulses in the analysis frame.

22. (Previously Presented) The method according to claim 19, wherein said convolution computation is calculated using the formula:

$$\hat{s}(n) = h(n) * u(n) = \sum_{k-1}^{F(n)} h(n - p(k))u(p(k))$$

wherein $\hat{s}(n)$ is the synthesized speech sample at time n, h(n) is the impulse response of the synthesis filter at time n, u(n) is the excitation function at time n, and p(k) is a location of the k-th excitation pulse in the frame; and

where said excitation function is defined by the formulas:

$u(p(k)) \neq 0 \quad \text{for} \quad k = 1,2 \ldots N_p$

$u(n) = 0 \quad \text{for} \quad n \neq p(k)$

and where F(n) is a number of excitation pulses in an analysis frame up to sample n and is defined by the formulas:

$p(F(n)) \leq n$

$F(n)) \leq N_p$, where N_p is the number of excitation pulses in the analysis frame.

23. (Original) The method according to claim 22, wherein said pulses are non-uniformly spaced.

24. (Original) The method according to claim 22, wherein said pulses are uniformly spaced; and wherein said excitation function is generated using a linear prediction coding ("LPC") encoder.

25. (Previously Presented) The method according to claim 22, further comprising a synthesis filter optimizer responsive to said excitation function and said synthesis filter and generating an optimized synthesized speech sample; wherein said synthesis filter optimizer minimizes a synthesis error between said original speech and said synthesized speech; wherein said synthesis filter optimizer comprises an iterative root optimization algorithm; and wherein said iterative root optimization algorithm uses the formula:

$$\partial\hat{s}(k)/\partial\lambda_r^{(j)} = b_r \sum_{m=1}^{F(k)} (k-p(m))u(p(m)) (\lambda_r^{(j)})^{(k-p(m)-1)}$$

where λ_r is the r-th root of the synthesis filter, $\lambda_r^{(j)}$ is the r-th root at the j-th iteration, and $\partial\hat{s}(k)/\partial\lambda_r^{(j)}$ is the partial derivative of the k-th synthesized speech sample relative to the r-th root of the synthesis filter at the j-th iteration.

APPENDIX B: Evidence

None.

APPENDIX C: Related Proceedings

None.